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HÖGSKOLAN KARLSKRONA RONNEBY

# Design and implementation of a test system for DECT audio measurement system

**Tore Nestenius**

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**Magisterprogrammet i elektroteknik**
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**Institutionen för signalbehandling**
**Examinator: Benny Lövström**
**Handledare: Patrik Holmberg, Flextronics VR/E**

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# 1 Abstract

This master thesis is for my exam in Master of Science in Electrical Engineering with emphasis on Telecommunications/Signal Processing at the university of Karlskrona / Ronneby. It was carried out at Flextronics International in Karlskrona and this report is a follow-up to my bachelor thesis made at the same premises. My assignment was to develop and implement an interface unit so that two different test instruments could communicate with each other. The instruments are used in the production of DECT (Digital Enhanced Cordless Telecommunications) based products. This report deals with the theory, design, construction and test of how to build this interface unit.

Result:
This report shows that it is possible to build a fully functional interface unit with few parts and that the quality of the interface box is as expected.

# 2 Introduction

## *2.1 Background*

The work in this thesis was carried out at the production facilities of Flextronics International Sweden AB in Karlskrona. Flextronics is a world leading contract manufacturer of electronic products and they have over 10.000 workers around the world.

One of the many products that Flextronics are manufacturing here in Karlskrona is the DRA-1900 system. DRA-1900 is manufactured for Ericsson Business Networks and it is based on the DECT (Digital Enhanced Cordless Telecommunications) standard. One part in the DRA-1900 system is the DECT Access Unit (DAU) and this thesis deals with the production tests of this product.

This thesis report is a direct follow-up to my bachelor report BEE 96-08 that i wrote in August 1996. That report was a pre-study of how to solve the problem and building a simple prototype. Flextronics has since that report was written changed DECT-tester test-instruments and that means that my previous report was no longer valid. This current report deals with the new conditions caused by the change of instrument and the building of a real prototype that can be directly used in live production.

## *2.2 DECT DRA-1900*

In a DECT radio access system, each subscriber has a fixed access unit (FAU) into which a standard telephone (or fax or modem) is plugged. The DECT Access Unit (DAU) is a part of the FAU. Somewhere nearby is a radio base station that serves all the radio-connected subscribers in the locality. A benefit of a DECT-based radio access system, compared to the use of systems based on one of the cellular standards, is that no frequency planning is required.

The DECT-based Ericsson DRA 1900 radio access system makes very efficient use of radio spectrum to link subscribers into the fixed telephone network, as an alternative or a complement to wired access.

The Ericsson DRA 1900 is a DECT-based WLL (Wireless Local Loop) system designed for use in urban, suburban and rural networks, to bring telephony and data services to residential and small business subscribers.

It is a cost-effective solution for small groups of subscribers, and can also serve high densities of subscribers (up to 100,000 users per square kilometre). Installation is faster than with wired access, and the installation or expansion of a DRA 1900 access systems requires no frequency planning.

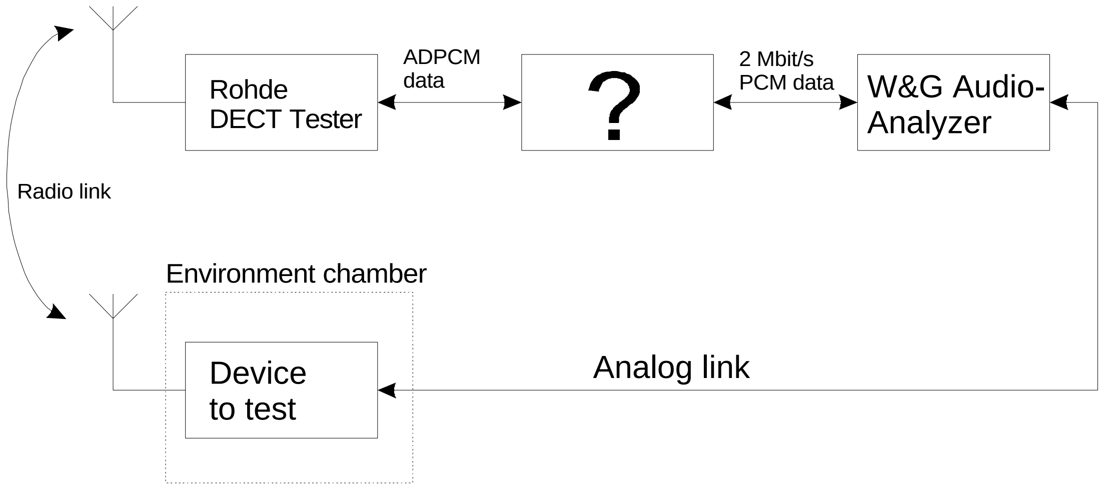
The essential element in this radio access system is the DECT radio base station, called the DECT Access Node. Measuring only 300mm x 500mm x 400mm, it can be installed on a rooftop or mast, and support up to approximately 600 radio-connected subscribers within a radius of a few kilometres. It is connected to the host local exchange by a 2 Mb/s wired link or an Ericsson MINI-LINK microwave link. Each subscriber is equipped with a DECT fixed access unit (FAU) into which a standard telephone is connected. Group 3 fax machines or data modems can also be connected. From the subscriber's viewpoint, having a DECT radio access to the telephone network rather than a conventional wired access makes no difference. The telephone socket is standard, the dialling and other tones are the same, and all the standard PSTN services are available. Speech quality is the same as with a wired access.

From the network operator's perspective, this digital radio access system provides a flexible way of adding new subscribers quickly and with minimal investment in civil engineering. Life cycle costs are low, and system planning and expansion is easy, since there is no frequency planning required.

Once the base station is installed in an area, extra subscribers can be added simply by installing the subscriber radio equipment on their premises. No hardware or software changes are required in the host local exchange when subscribers are connected by DRA1900. Even the PSTN numbering plans are unaffected.

### *2.3 The test-configuration*

The configuration of the involved test instrument and test object is shown in the figure below. Due to the fact that the two test-instruments can’t communicate directly with each other, some sort of translating device had to be built and the “?” box below shows the device that this report is all about.



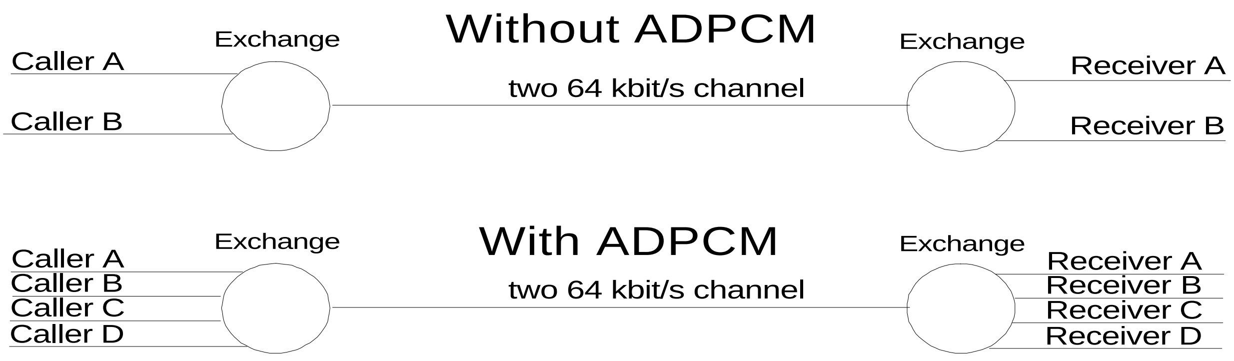
# 3 ADPCM

ADPCM is a speech compression algorithm that is used in the DRA-1900 system to reduce the number of bits needed to represent the speech data in each channel. To be able to build this converter, some basic knowledge of the ADPCM compression algorithm is necessary and in this chapter, we will take a deep look at how the ADPCM compression algorithm operates.

Adaptive differential pulse code modulation (ADPCM) compression is a very efficient way to code digital waveforms. It is widely used in speech compression application because it makes it possible to reduce the bit flow while maintaining an acceptable level of quality. ADPCM is also used for other types of data like data-storage, image, high-quality audio and modem data.

The use of ADPCM compression in, for example telecommunication exchanges can double the number of callers per trunk and therefore double the profit for the operator. We can now send two 32 kbit/s ADPCM voice connections over a standard 64 kbit/s connection instead of just sending one connection. It is also quite easy and cheap to implement ADPCM compression in the existing telephone system.

The picture below shows the benefits of using ADPCM compression in the telephone system. Instead of just being able to send two calls on two 64kbit/s channels, we can now with the help of ADPCM send four calls at the same time over the same channels.




The basic principle of ADPCM is to use our knowledge of the signal in the past time to predict it in the future and the resulting compressed signal is the error of this prediction.

Many different ADPCM algorithms have been proposed and we are in this report only dealing with the 32 kbit/s ADPCM recommendation G.726 from CCITT that is widely used in the telecommunication world.

The CCITT (The International Telegraph and Telephone Consultative Committee) is a permanent organ of the International Telecommunication Union (ITU). They are responsible for studying technical, operating and tariff question and issuing recommendations on them with a view to standardizing telecommunications on a worldwide basis.

Implementing ADPCM compression can be done in either software or hardware and in this report we are using a single ADPCM hardware chip that handles all the conversion. G.726 has an effective bandwidth of 4Khz when operating at 32kbit/s datarate. One advantage with G.726 ADPCM compression compared with other algorithms is the low signal delay.

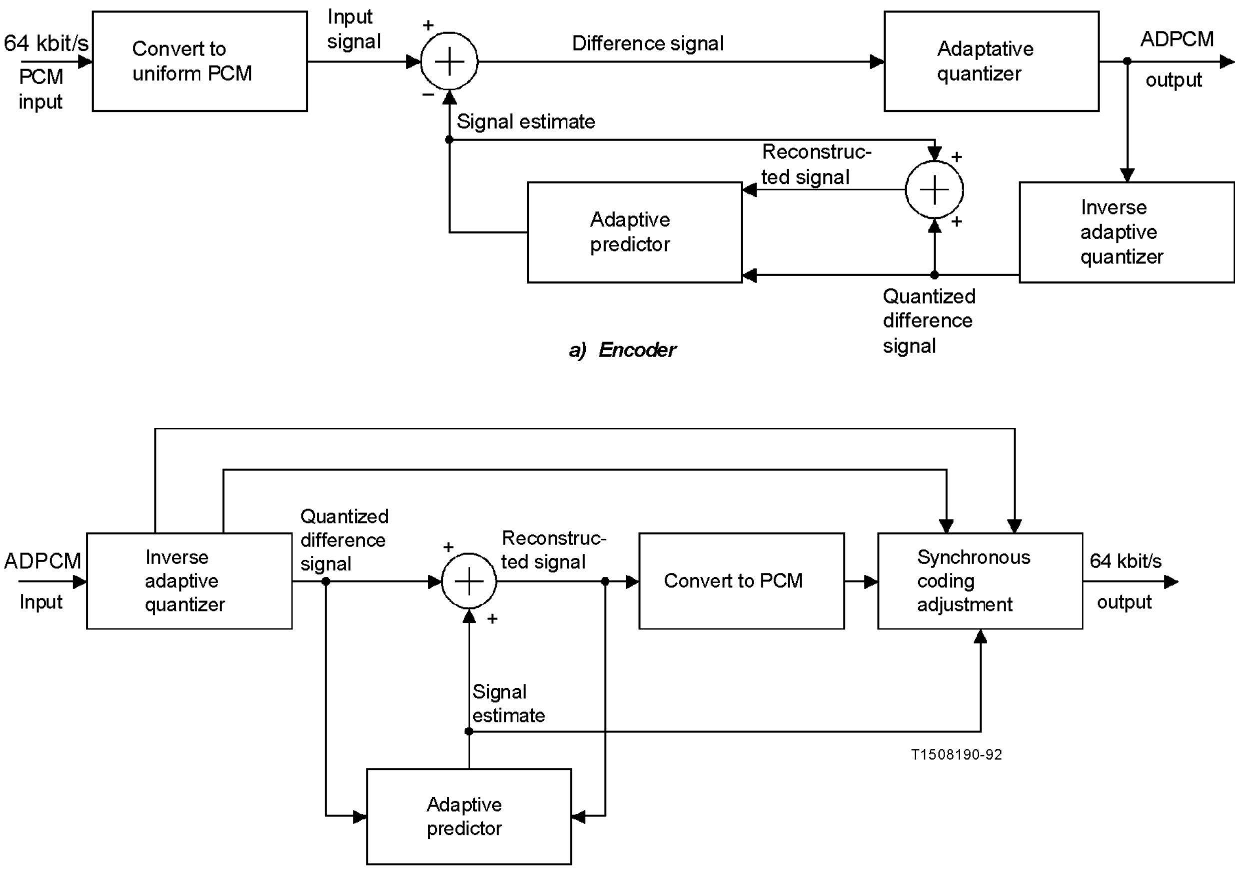
### *3.1 ADPCM recommendation G.726*

This recommendation deals with the conversion of a 64 kbit/s A-Law or µ-Law pulse code modulation (PCM) channel to and from a 40, 32, 24 or 16 kbit/s channel. The conversion is applied to the PCM bit stream using an ADPCM transcoding technique. Note that we are only referring to the 32 Kbit/s ADPCM version in this report.

The CCITT ADPCM was designed to meet several specific requirements, the design goals were to:

- Provide compression while satisfying the objective signal quality requirements specified in the CCITT recommendation G.712.
- Have a high enough level of subjective quality (defined by listening tests) even after a series of encodings and decodings.
- Provide compatibility with the existing A-Law and µ-Law PCM formats.
- Operate stable in the presence of high bit-error rates during transmissions.
- Operate properly on the presence of voiceband data (modems) at rates up to 4.8 Kbit/s.

The block diagrams below shows a simplified version of the ADPCM compression encoder and decoder:

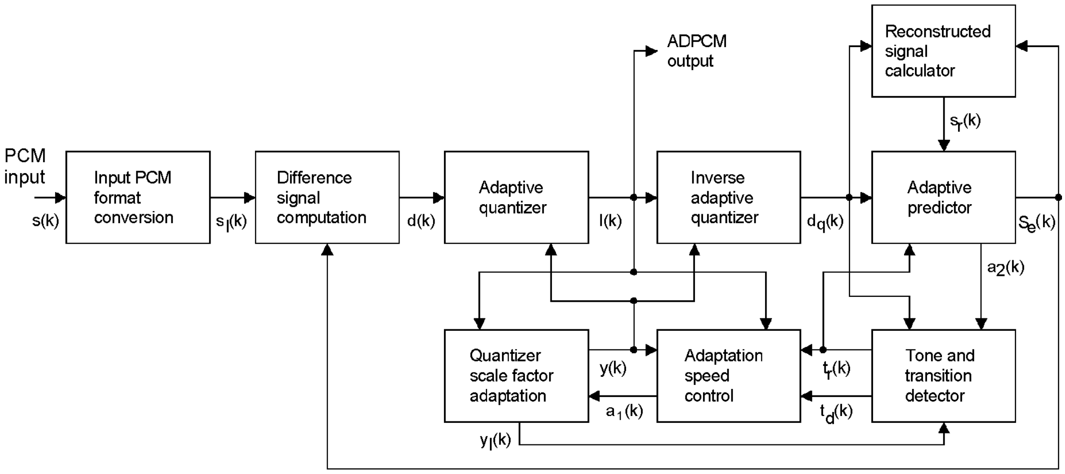



*a) Encoder*

*b) Decoder*

### *ADPCM Encoder*

The purpose of the ADPCM encoder is to convert the 8-Bit PCM A-Law or µ-Law data to 4-bit ADPCM data. The block schematic below shows the structure of the ADPCM encoder:

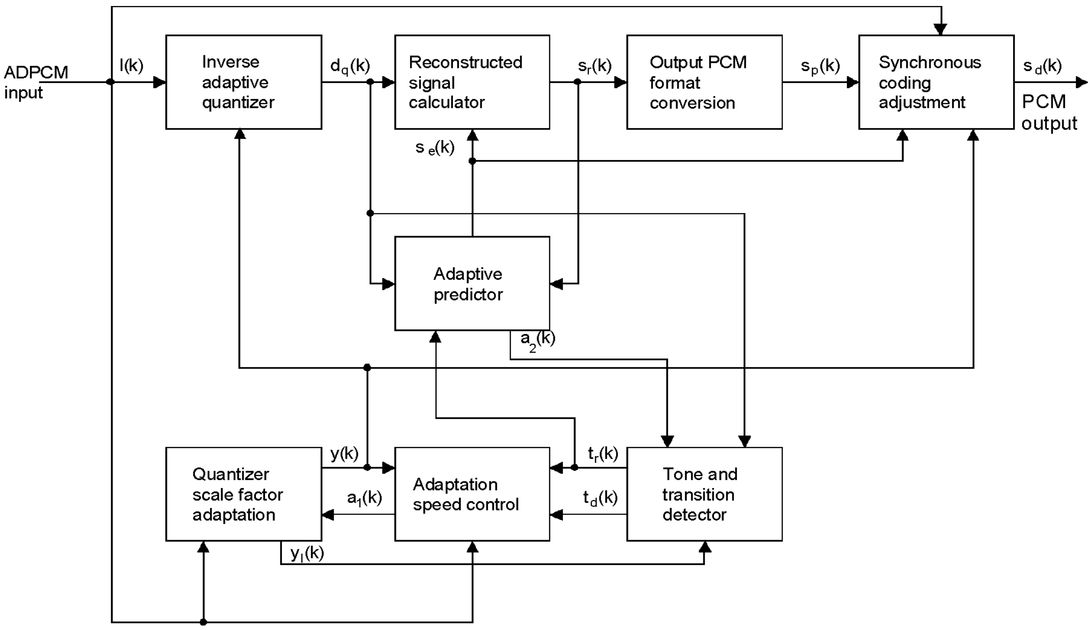



The encoder starts with a conversion of the A-Law or µ-Law PCM input signal to uniform PCM. After that a difference signal is obtained, by subtracting an estimate of the input signal from the input signal itself. An adaptive 15 level quantizer is then used to assign 4 binary digits to the value of the difference signal. The resulting 4 bits are then send to the output of the encoder and to the adaptive predictor.

An inverse quantizer produces a quantized difference signal from the output of the encoder. The signal estimate is added to this quantized difference signal to produce the reconstructed version of the input signal. Both the reconstructed signal and the quantized difference signal are fed into an adaptive predictor who produces the estimate of the input signal, thereby completing the feedback loop.

### *3.2 ADPCM Decoder*

The purpose of the ADPCM Decoder is to convert the 4-bit ADPCM data to 8-Bit PCM A-Law or μ-Law data. The decoder includes a structure identical to the feedback portion of the encoder, together with a uniform PCM to A-Law or μ-Law conversion and a synchronous coding adjustment unit. The synchronous coding adjustment prevents cumulative distortion occurring on synchronous tandem codings (ADPCM-PCM-ADPCM, etc. digital connections),




### *3.3 The ADPCM internals*

As you can se above, ADPCM contains several subparts and they are all described below:

#### 3.3.1 Encoder parts

- Input PCM format conversion
  This block converts the input signal from A-Law or μ-Law PCM to a uniform PCM signal.

- Difference signal computation
  Calculated the difference signal from the uniform PCM signal and the signal estimate.

- Adaptive quantizer
  This is a non-uniform adaptive quantizer that is used to quantize the difference signal into 15 discrete levels. Prior to quantization, the signal is converted to a base 2 logarithmic representation and scaled by a factor from the scale and speed factor adaptation block. The reason for quantizing in the logarithmic domain is becuase a more uniform signal-to-noise ratio can be achived.

- Inverse adaptive quantizer
  Here the signal is transformed back from the quantized logarithmic domain.

- Quantizer scale factor adaptation
  This block computes the scaling factor for the adaptive quantizer and inverse quantizer.

- Adaptation speed control
  Here a control parameter is generated that tells the scale factor adaptation block the rate-

of-change of the signal. The rate-of-change speed is used to tell if the input signal is a slow speech signal or faster voiceband data signal. The speed-control factor weights the fast and the slow adaptation factors for form a single quantization scale factor.

- Adaptive Predictor and Reconstructed signal calculator
  This block is responsible for calculating the signal estimate that is used by the difference signal computation block described above.

- Tone and transition detector
  This block is used to improve the performance for signals originating from frequency shift keying (FSK) modems

### 3.3.2 Decoder parts

- Output PCM format conversion
  This block converts the reconstructed uniform PCM signal into an A-Law or μ-Law PCM signal.

- Synchronous coding adjustment
  This block prevents cumulative distortion occurring on synchronous tandem coding PCM-ADPCM-PCM-ADPCM-PCM etc. This works when the transmission channel is error free and when external digital signal processing does not disturb the ADPCM data.

Se the encoder parts for description on the rest of the blocks.

## *3.4 Adaptive predictor*

Two adaptive predictors are used, as a sixth order section that models zeros and a second order section that models poles in the input signal. This dual structure effectively caters for the variety of input signals that might be encountered. The adaptation method used is the feedback type and this means that no extra control information has to be transmitted with the data. The predictor coefficients are updated using a simplified gradient algorithm. The sixth-order all-zero section helps to stabilise the filter and to prevent it from drifting into oscillation. The predictor coefficients are updated on every sample by a simplified gradient search algorithm.

## *3.5 Verification of the ADPCM encoder and decoder*

The verification of ADPCM is very difficult to perform due to the adaptive nature of the ADPCM algorithm. This means that there is almost an infinite number of possible test sequences.

To avoid having to test all possible test sequences CCITT have pre-generated digital test sequences that are used to verify the conformance of an ADPCM implementation. These sequences have been chosen to exercise the major arithmetic parts and thus giving a reasonable level of confidence of the implementation. Special reset-sequences are used before for every test to reset the ADPCM converter into a default know state. Otherwise, the results would not always be the same.

The test sequences consists of 39 different test sequences and they divided in the following three categories:

1. A general set of various sinusoidal input PCM signals
2. Overload vectors, consisting of very high level input PCM signals
3. Special sequences that exercise the ADPCM algorithm in a manner that is not possible with a normal PCM input sequence.

### *3.6 ADPCM performance*

The performances of ADPCM channels are very signal dependent and random signals like data transmissions should be avoided. ADPCM compressed links are not suitable for high speed data transmissions and only transmissions up to 2400 bit/s can be used without serious degradation of the transmission speed.

ADPCM systems are affected differently by bit errors than PCM systems because the ADPCM decoder loop causes error propagation, while a PCM error does not propagate in time.

Under error free transmission conditions the perceived quality of transmitted speech over 32 kbit/s ADPCM links is only slightly lower than that over 64 kbit/s PCM links. However, when the transmission error ratio is higher than $10^4$ bits/error the perceived quality of speech over 32 kbit/s ADPCM links is better than that over 64 kbit/s PCM links.

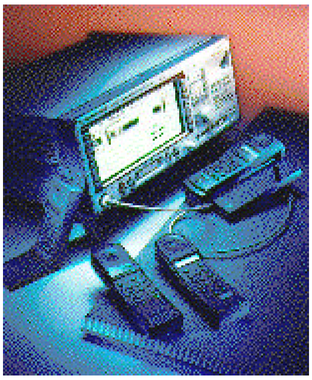
# 4 Rohde & Schwarz Digital Radio Communication Tester

The Rohde & Schwarz Digital Radio Communication Tester CMD-60 is a flexible instrument that is used when performing tests on DECT compatible devices. The instrument is used at Flextronics in the testing of the customers DECT products.

The CMD-60 instrument can for example test the following parameters on a DECT-device:

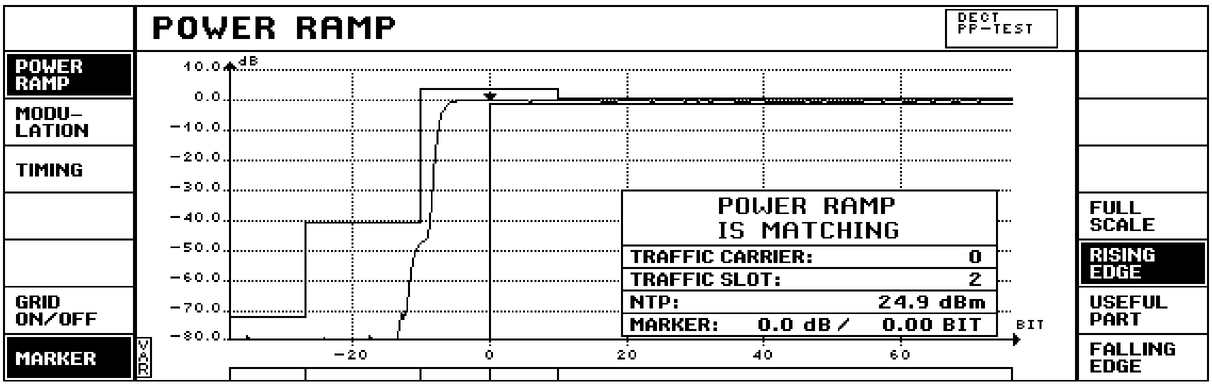
- Accuracy and stability of the RF-carrier
- Timing jitter and accuracy
- Reference timing accuracy of a RFP (Radio Fixed Part, a DECT-device)
- Packet accuracy and delay
- Transmission burst
- Transmitted power
- RF carrier modulation
- Radio receiver reference sensitivity
- frame and Bit error

The CMD-60 instrument looks like this:

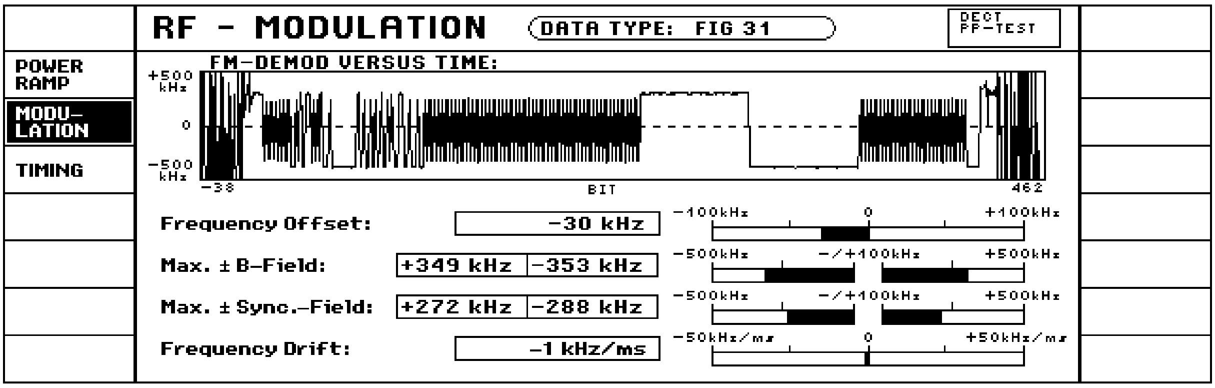


## 4.1 Typical displays of CMD 60

Here are two examples of what the CMD-60 looks like when making power-ramp and RF-modulation measurements.




Power ramp measurement gives in-depth analysis of the burst power transmitted by the DECT unit. The measurement is synchronized to bit P0, thus giving precise information not only about the power transmitted but also about timing parameters.




The RF modulation measurement menu presents in a scope picture the demodulated signal for easy and quick recognition of typical data forms.

In this project the CMD-60 have the following purposes:

1. Establish and maintain a reliable RF connection between the CMD-60 and DAU while the test lasts.

2. Act as a transparent bridge that passes through the ADPCM voice data that is sent between the DAU and the CMD-60 to the external ADPCM-port. The translation device will then translate the ADPCM data to PCM-format and then feed it to the W&G instrument.

3. Generate the 8KHz sync that is used to synchronize the system.

### *4.2 CMD-60 ADPCM interface*

On a standard CMD-60, it is possible to connect external devices that can listen to the ADPCM data sent from the DAU to the CMD-60. The data can be extracted from the 50-pin multi-connector that can be found on the rear of the CMD-60 instrument. To allow external devices to send data to the DAU via the CMD-60 some sort of internal modification has to be made to the CMD-60. The modification is made by Rohde & Schwarz themself to avoid any warranty and calibration problems. For this project, Rohde & Schwarz have modified one instrument and it is now possible to have full control of the ADPCM data send to and from the DAU.

The CMD-60 must be configured to activate the external ADPCM-input pin. This has to be done every time the CMD-60 is turned-on or reseted. The activation can be done in the following ways:

1. Manually, by using the following special undocumented escape sequences in the ”connection Established” menu.

   ”EE00000000”, Disables the external ADPCM input pin.
   ”EE01000000”, Enables the external ADPCM input pin.

2. By software, by sending the following special undocumented IEEE-488 commands:

   SEND:DECT:COMMAND ”100E2008002000”  Enables external ADPCM input
   SEND:DECT:COMMAND ”100E2008000020”  disables external ADPCM input

The ADPCM-interface consists of the following signals:

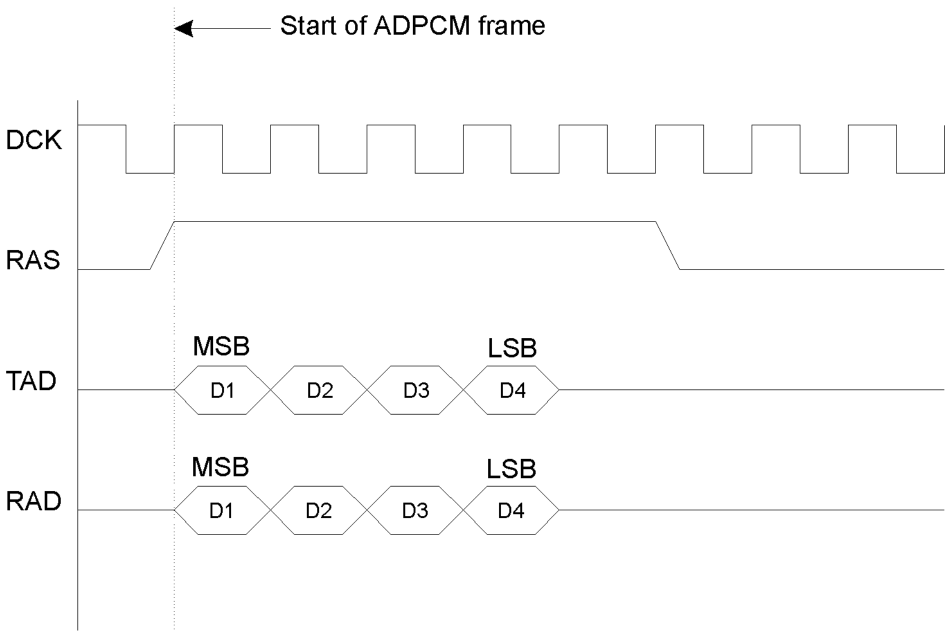
- DCK (Digital Clock, output), this is the 1.152MHz clock signal that defines the transfer rate of the data.

- RAS (Receive ADPCM Strobe, output), 8kHz synchronization signal, that defines the start of the data transfer. This sync is used for synchronizing this project. The PCM-5 bases all its operation on this signal. The width of this signal is about 10µs.

- TAD (Transmitt ADPCM Data, input), the CMD-60 reads the serial ADPCM data on this pin. On the rising edge of the RAS signal, the CMD-60 will start reading 4 ADPCM bits on the 4 first high-to-low transitions of the DCK data clock.

- RAD (Receive ADPCM Data, output), the CMD-60 outputs the serial ADPCM data on this pin. On the rising edge of the RAS signal, the CMD-60 will start transmit 4 ADPCM bits on the four first low-to-high transitions of the DCK data clock. The data is actually sent out 12 times in each 8KHz time slot because there are 12 timeslots in the DECT standard. In this project, all data after the four ADPCM bits are ignored.

The figure below shows the timing and format of the above ADPCM signals:




Timing diagram for the CMD-60 ADPCM interface

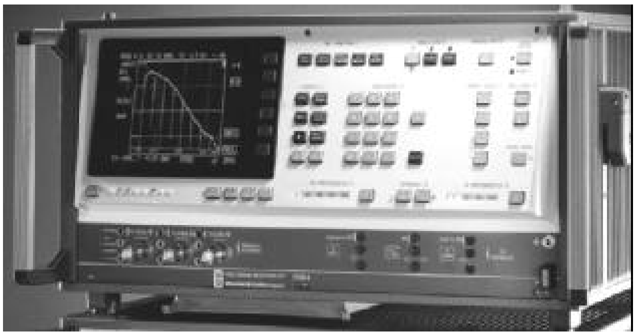
See the Philips PCD5032 CODEC manual for more information about the ADPCM timing and format that is used by the CMD-60.

# 5 Wandel & Goltermann PCM Channel measuring set

The Wandel & Goltermann PCM Channel measuring set PCM-5 is a general-purpose telecommunication test instrument that is used in this project for making the actual audio measurements of the DAU.

The PCM-5 are reference measuring sets which can be used to determine characteristics between the analog and digital levels ("half-channel" measurements) with excellent speed and precision. The instruments include built-in self-tests and calibration routines to handle the demanding requirements that the digital systems used in transmission and switching makes of test and measurement equipment.

The instrument looks like this:



The PCM-5 can for example perform the following measurements with a high precision:

- Level measurements
- Overall loss, echo return loss
- Variation of gain with frequency
- Variation of gain with input level
- Total and harmonic distortion
- Idle channel noise
- Crosstalk
- Overload capacity
- Digital error measurements and frame evaluations

## *5.1 Digital Interface:*

To communicate with the W&G instrument, we are using the standard E1 2.048Mbit/s PCM-interface. This type of interface is very common in telecommunication environments and it is normally used between telephone exchanges.

The 2.048 Mbit/s PCM (Pulse Code Modulation) communication standard is defined by CCITT and can transmit 30 telephone or data channels in each direction over a digital 2.048 Mbit/s channel. This type of channel is also called E1 and used mainly in Europe and some non-European countries.

Each E1 2.048 Mbit/s channel carries 30 voice/data plus two auxiliary channels. The two auxiliary channels are used for framing, synchronization and signaling between the transmitter and receiver. Everything sent on the E1 channel is grouped in frames. Each frame contains an 8-bit sample of each channel and the transmitter sends 8000 frames every second. 8-Bits per frame and 8000 frames a second generates an effective data rate of 64 Kbit/s.

When voice data is sent over a channel, a compression standard called A-Law is applied to the signal. The A-Law standard is defined in the ITU-T recommendation G. 711. For more detailed information about the A-Law and PCM, see the CCITT/ITU-T standards. Knowledge of the PCM format and nomenclature of the E1 operating environment are necessary when constructing around this communication system.

A PCM channel can be configured in many different ways and in this project, we are using the following settings:

- A-Law compressed data.
- HDB-3 digital transmission line encoding.
- 75-Ohm coaxial transmission cables.
- Timeslot 0 and 16 are used for synchronizing and signaling.
- W&G instrument is externally synchronized.
- Timeslot 1 is used for data transmission, all other slots are unused.
- 2.048MBit/s bitrate

# 6 The translator device

This chapter has been removed from this report because it contains classified information.

# 7 PCM measurements

Several transmission measurements are used when measuring the transmission characteristics of a pulse code modulation (PCM) channel. How to perform these measurements has been standardised by the CCITT (the international telegraph and telephone consultative committee).

The PCM channel-measuring instrument from Wandel & Goltermann that is used in this project can perform most of these measurements. More detailed information can be read in the G.712 and G.714 standard from CCITT and the W&G operating manual.

This chapter will deal only with the measurements that are used in this project.

## *7.1 PCM Level measurements*

Here the received level of the sent signal is measured and by using this test it is possible to get an overall view of what happens to the signal level as it passes through the system.

## *7.2 Overall loss*

This loss measurement measures the level difference between the expected and the actually received level.

This measurement is performed by first internally connecting the signal generator directly to the receiver. The received level is measured and after that the internal loop is opened and a second measurement is made through the external test object. The result is the difference in level between the internal measurement and the external measurement. The reason for doing this is that the attenuation characteristics of the generator and receiver should not influence the result.

## *7.3 Variation of gain with frequency*

This measurement shows the variation of gain at different frequencies. First a reference level is measured at a fixed reference frequency. Then a single frequency sweep is performed and the result is the difference between the reference level and the level measured at the different frequencies.

## *7.4 Variation of gain with input level*

With this measurement, it is possible to measure linearity of the test-object. First a reference level is measured at a fixed reference level and frequency. Then a single level sweep is performed at the same frequency and the result is the difference between the reference level and the level measured at the different levels.

## *7.5 Total distortion including quantization distortion*

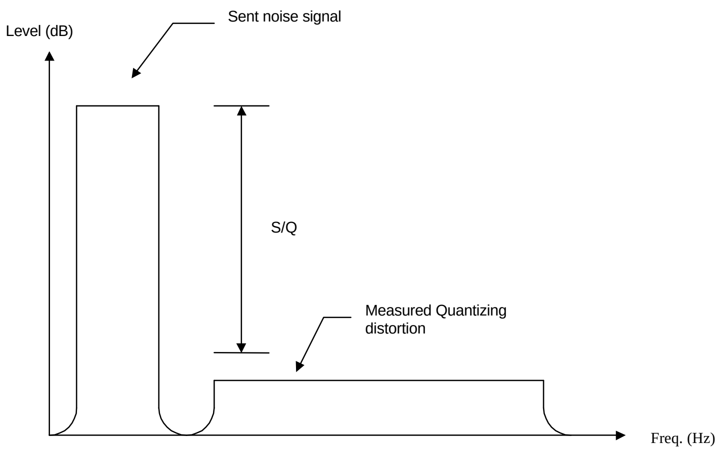
Here the total distortion and the quantization distortion of the system are measured. Quantization distortion occurs for example when a:

- Analog signal is sampled and described with discrete values.
- Digital linear signal is compressed/decompressed with A-Law or μ-Law
- Digital PCM signal is compressed/decompressed with ADPCM

There are two methods used when measuring the quantizing distortion:

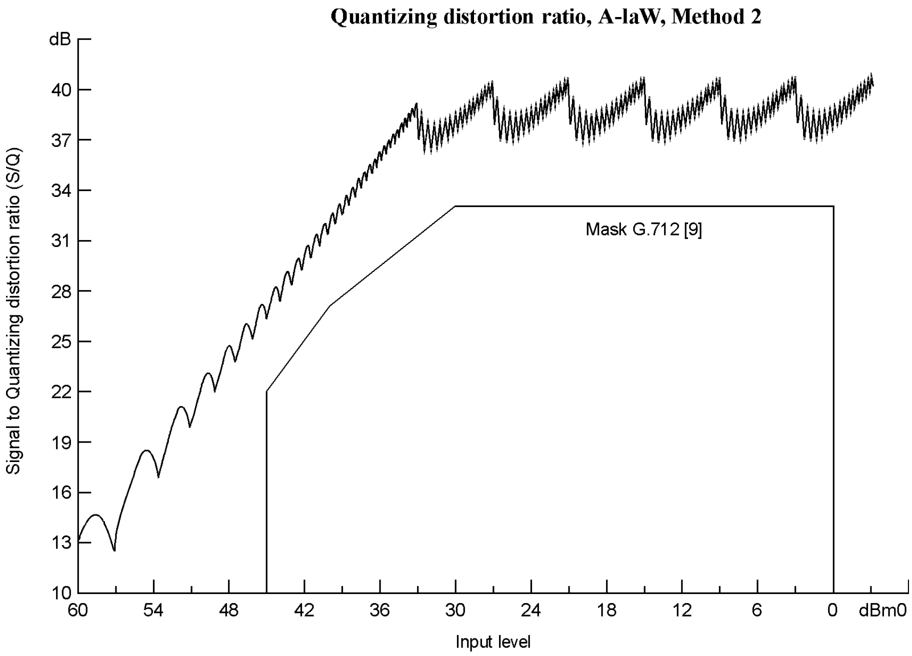
1. A band-limited noise signal with a power-spectrum within the range 350 Hz to 550 Hz. is applied to the input port of a channel. The signal-to-total distortion power is then measured as the ratio of the power in the frequency range 350-550Hz and in the 800-3400 Hz range. The noise in the 800-3400Hz range is caused by the quantizing distortion.
2. Here the noise signal is replaced with a sinusoidal signal with the frequency 1020 Hz.

The picture below shows the principles for the measurement:




The resulting distortion is measured as the level difference between the test signal and the received quantized signal.

The diagram below shows the results of an quantizing distortion measurement on a error-free PCM channel. The ripple in the graph is caused by the quantizing noise and the A-Law compression.



# 8 Evaluations

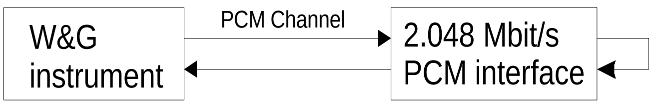
The resulting measurements in this chapter have been removed because they contain classified information.

Several measurements were done on the device to verify that the device operated as expected. All the tests were performed using the W&G instrument that were acting as a signal generator and a signal analyser at the same time. The result of the tests was very satisfying and the results were checked and confirmed by one of Flextronics own measurement experts.

See the PCM measurement chapter for information about the various PCM-measurements that are described in this chapter.

## *8.1 Evaluation of the PCM interface using bit-error tests*

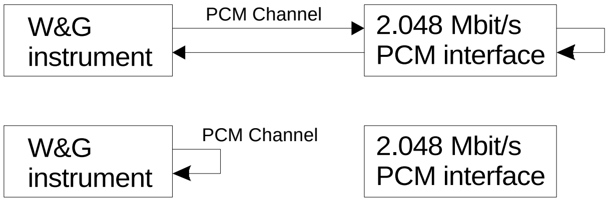
To test the 2.048 Mbit/s PCM interface we used the bit-error test function on the W&G instrument. The W&G instrument can perform various kinds of bit-error measurements and in this test, we used the pseudo random bit-generator to generate the test-sequences. The schematic below shows the configuration of the bit-error test.




After running this test for over 30 minutes we found that no bit errors were detected and this means that we can be quite confident that there’s no interfering distortions or bit-slips. Performing a PCM->ADPCM->PCM bit-error test is not possible due to the destructiveness by the ADPCM compression.

## *8.2 Evaluation of the PCM interface using audio tests*

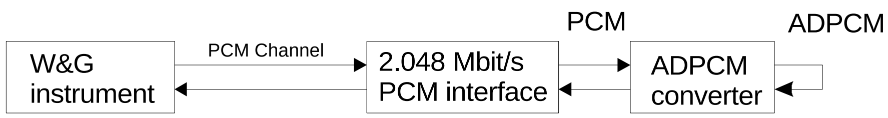
To ensure that there were no audio-distortions on the PCM-part of the design we compared the results of various audio-measurements of the two configurations below.




No difference in the two above configurations was detected and now when the bit-error and audio tests have passed, we can be very confident that we have a reliable PCM-communication.

## 8.3 Evaluation of the ADPCM compression using audio tests

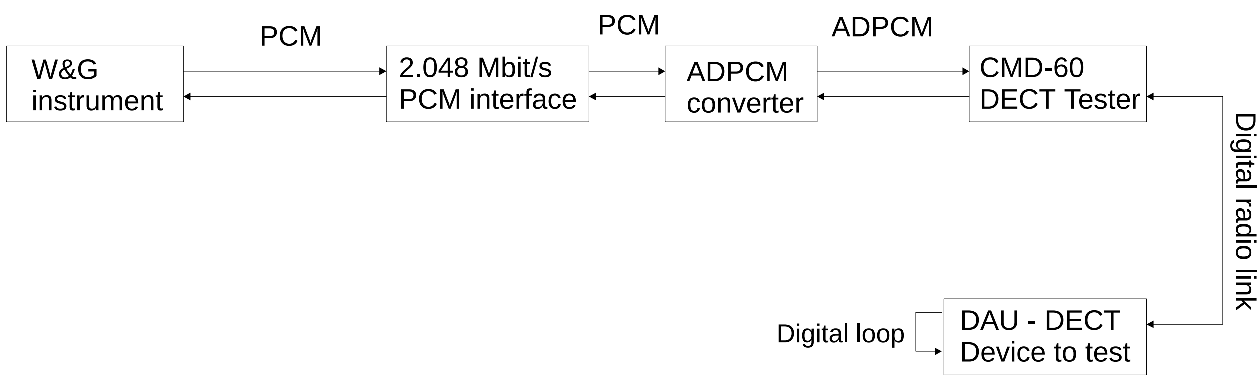
The following test configuration was used in this test:




To evaluate the performance of the ADPCM converter we performed the same tests as we did above when we tested the PCM channel. The result shows that the performance is dropping a little due to the ADPCM converter, but it is still within acceptable limits.

## 8.4 Evaluation of all digital connections using audio tests

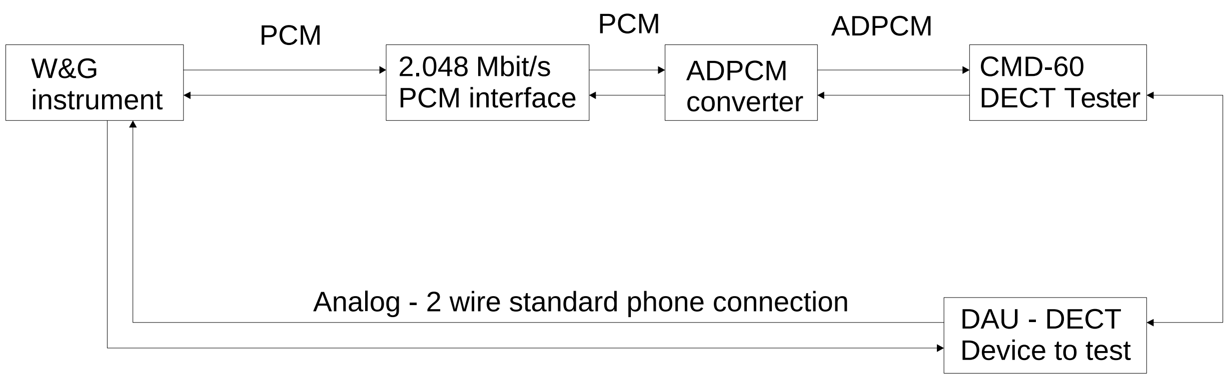
To test all the digital connections (including the radio and DAU) we used the same audio tests above and the following configuration:




The DAU was patched inside to enable a digital loopback of the digital data to/from the DECT-tester. The results were as expected exactly the same as the previous digital tests.

## 8.5 Evaluation of the complete design using audio tests

To evaluate the complete system including DECT-tester, DAU (unit to test), ADPCM and W&G we used the following configuration:



The signal path in this configuration includes PCM, ADPCM and analog connections, this means that there are many sources of errors when performing these measurements.  In this test we are using two different measurement modes, Digital->Analog and Analog->Digital tests. These modes represent the two different measurement directions that can be measured in a DECT system.

The result shows that measurements in both directions are matching well, except at the lower frequencies/levels where there is some difference. This is due to the quantization and compression of the signals.

# 9 Conclusion

I only had one major problem in developing this product and that was the lack of accurate documentation on the CMD-60 ADPCM interface. I spent over 5 weeks investigating why the ADPCM data received by the CMD-60 was corrupted by noise. During this time, i reconfigured the design over 10 times and it was very frustrating not being able to solve the problem.

I found the solution on my last week at Flextronics and it turned out that there was an undocumented inverting schmittrigger circuit on the ADPCM input stage. This meant that I had to externally invert the ADPCM data send to the CMD-60. After adding this inverter, everything worked fine and as planned. I found this bug by analysing the digital data stream sent over the radio.

I learned my lesson that i should never trust written documentation, since it actually can contain serious errors.

After solving this problem, I finally had a clean, reliable and simple design that did exactly as planned. The result of the tests was very satisfying and the results were checked and confirmed by one of Flextronics own measurement experts. Flextronics now have the solution to their measurement problem and they can now start testing the DAU-product the way they wanted to in the first place.

# 10 References

## *10.1 Databooks and manuals*


- Design and evaluation of an interface for DECT audio testing, bachelor thesis number BEE 96-08 by Tore Nestenius, University of Karlskrona/Ronneby , August 1996.

- Object specifications, ADPCM CODEC for Digital Cordless Telephone PCD5032 from Philips semiconductors.

- Wandel & Goltermann PCM-5 operating manual, BN 0984/00.71

- Operating manual for CMD60 digital radiocommunication tester for DECT 1050.9008.60 from Rohde & Schwarz

- Telecommunications Data Book from Dallas Semiconductor. (DS2164Q)

- DSP56000, Digital signal Processor family manual, DSP56KFAMUM/AD

- DSP56002, Digital Signal Processor user’s manual, DSP56002UM/AD rev1

- Digital Signal Processing using the Motorola DSP Family written by Robert J. Simpson, ISBN 0-13-490632-2, Prentice Hall

- Motorola Semiconductors, DSP Application Notes

- DRA 1900, System manual, LZB 133 130 P1C, by Ericsson Business Network AB

- Digital Coding of Waveforms, principles and Applications to speech and video written by N.S. Jayant - Peter Noll, ISBN 0-13-211913-7

- Advanced electronic communications systems written by Wayne, Tomasi, Prentice Hall, ISBN 0-13-288309-0

- Digital Transmission, Kenth Lindgren, Liber utbildning, ISBN 91-634-0049-9

### 10.2 WWW references

| | |
|---|---|
| http://www.programmersheaven.com | Programmers Heaven |
| http://www.motorola.com | Motorola |
| http://www.hp.com | Hewlett Packard |
| http://www.wg.com | Wandel & Goltermann, Inc. |
| http://www.dalsemi.com | Dallas Semiconductors |
| http://htsa.htsa.hva.nl/~arjanj/dect.htm | DECT homepage |
| http://www.mitel.com | Mitel |
| http://www.eg3.com/dsp.htm | DSP Internet Resource List |
| http://www.analog.com | Analog Devices |
| http://www.burr-brown.com | Burr-Brown Corporation |
| http://www.natinst.com | National Instruments |
| http://www.tapr.org/dsp | DSP Experimenter's Page |

### 10.3 Standards & recommendations

- ITU-T recommendations G. 701, Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms.

- ITU-T recommendations G. 703, Physical/Electrical characteristics of hierarchical digital interfaces.

- ITU-T recommendations G. 711, Pulse Code Modulation (PCM) of voice frequencies (A-Law)

- ITU-T recommendations G. 712, Transmission performance characteristics of pulse code modulation.

- ITU-T recommendations G. 726, 40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code modulation (ADPCM)

- Technical Basis for Regulation (TBR) 10, DECT general terminal attachment requirements: Telephony applications.

- European Telecommunication Standards (ETS) 300 175-8, Digital European Telecommunications (DECT) Common Interface (CI); Part 8: speech coding and transmission.

# 11 Component list

## *11.1 Component list for this project*

### 11.1.1 Resistors:

| | |
|---|---|
| R1, R2, R3, R4 | 330 Ohm |
| R5, R6, R7, R8 | 270 Ohm |
| R9-R13, R17 | 4.7 Kohm |
| R14-R16 | 220 Ohm |

### 11.1.2 Capacitors:

| | |
|---|---|
| C1, C2, C3, C4, C5 | 0.1uF |
| C6, C7 | 10uF |
| Decoupling capacitors not listed | |

### 11.1.3 Integrated Circuits:

| | |
|---|---|
| IC1, IC2, IC3, IC4 | HP2400 optocouplers |
| IC5, IC6 | 74HCT04 |
| IC7 | DS2164Q (ADPCM Transcoder) |
| IC8 | 74HC241 Buffer |

### 11.1.4 Various components

| | |
|---|---|
| D1-D2 | Greed LED |
| D3 | Red LED |
| B1 | Dallas DS2153Q Evaluation board |

## 12 Schematics

This chapter has been removed from this report because it contains classified information.